X6v VoIP Features

T E C H N I C A L   R E F E R E N C E

Contents

1 Introduction

This document describes the ADSL X6v modem's VoIP features. It provides information about the VoIP configuration parameters and explains how to view and modify them using the Configuration Manager interface or by downloading configuration files via the VoIP Subsystem's update mechanism.

Using the Configuration Manager

The **Configuration Manager** is the interface to the ADSL X6v modem. To access the interface:

1 Type http://192.168.0.1 in your browser's address field.

2 When prompted, log on in administrator mode, using the following Username and Password:

Username: **admin**
Password: **zoomadsl**

Note to service providers: If you are going to lock units to your service, we strongly recommend that you change the password before shipping any product to the field. See Changing the admin Password on page 6 for instructions.

3 When the **ADSL Setup** page opens, click the **VoIP** icon on the Zoom menu bar to access the VoIP Subsystem.

4 Click the **Advanced VoIP Setup** icon, then select **VoIP System** from the left pane's menu to access the configuration parameter categories.

5 Select items from the **VoIP System** menu to view or modify the parameters within these groups:

- System Parameters
- VoIP Accounts
- VoIP Parameters
- SIP Parameters
- Regionalization
- Subscription Services
- User Configuration
- Feature Codes

Downloading Configuration Files

Configuration files are prepared and stored on the service provider's update server. At power up, reboot, or configurable periodic intervals, the VoIP Subsystem can contact an update server. When it contacts the update server, the VoIP Subsystem provides unique identification. The update server then checks a database to determine whether there is new firmware and/or a configuration file for the VoIP Subsystem. If there is, the update server instructs the VoIP Subsystem to download the relevant file or files. The configuration server can use the VoIP Subsystem's device identification to prepare a specific configuration file that might include, for example, detailed account information.

Changing the admin Password

To change the admin password:

1. Type **http://192.168.0.1** in your browser's address field.
2. When prompted, log on in administrator mode:

   Username: **admin**
   Password: **zoomadsl**

3. When the **ADSL Setup** page opens, click the **Router Setup** icon on the Zoom menu bar.
4. On the **Router Setup** page, click **Admin Password**.
5. On the **Admin Password Configuration** page, type the old and new passwords, then confirm the change.
6. Click **Save**.
7. When the authentication dialog opens, type the new password in the **Password** field, then click **OK.**
8. Click **Write Settings to Flash**.

Important!

If you change the **admin** password, and then forget the new password, you cannot retrieve it. You will need to reset the unit to the factory default settings which will erase any previously saved (changed) settings.

2
Changing Configuration Parameters

As administrator (**admin**), you can view and modify the VoIP configuration parameter values described in this *Technical Reference* and set user access privileges for each parameter. See Chapters 3 through 10 for a description of the available menus and configurable parameters.

Setting User Access Privileges

When you are logged on as **admin**, the VoIP interface displays a pull-down menu labeled **User** to the right of each configurable parameter. The pull-down menu values are **E**, **P**, **V** and **-** (dash). The value that you choose defines user access privileges for each field.

Value	Description
E	Full Edit capabilities. Read, write, delete.
P	Full Edit with Priority. Cannot be overwritten by config download via update server
V	View. Read only.
-	No access. (This value is not seen by the user.)

Note: Each account page has only one pull-down menu that controls access for all fields on that page. On some pages, there are additional pull-down menus to the right of the User fields. These menus are labeled Phone and they control access to features (setting up speed dials, call blocking based on caller ID, etc.) that can be activated using a handset. For the Phone pull-downs, only the symbols **E** and **-** (dash) are available.

3
System Parameters

You can use the **VoIP -> Advanced VoIP Setup -> VoIP System** menu to configure overall system settings. The menu items include:

- VoIP System Identification
- Date/Time
- VoIP Subsystem Network Configuration
- Static IP/DNS Configuration
- HTTP / Telnet / FTP Server
- STUN Settings
- Firmware and Configuration Update Settings
- VoIP System Maintenance

VoIP System Identification

Parameter	Description	Default
Boot ROM Revision	Boot code revision	6.3.1
Firmware Revision	Run-time code revision	6.3.1
Configuration Revision	Configuration file revision	6.3.1 – 00/70/72
MAC Address	Ethernet MAC address assigned during manufacture	(as assigned)

Note: Default revisions will vary according to the release date of your product. Configuration suffixes vary by region

Date/Time

Parameter	Description	Default
Date (yyyy/mm/dd)	Current date	
Time (23:59:59)	Current time	
Time Zone (rel. GMT; -12 to 13)	Number of hours to subtract from GMT to form local time	-5
Daylight Savings	Enable or disable local application of daylight savings time	Enabled
Obtain Time from NTP Timeserver	Enable or disable use of network timeserver	Enabled

Parameter	Description	Default
NTP IP Address	Fully qualified domain name (FQDN) (including an optional port number) for the NTP/SNTP timeserver server	time-a.nist.gov

Note: When the date and time are set independently of NTP (that is, if a timeserver is unavailable or the use of a timeserver is disabled), adjustments must be made to a time at least one hour ahead or behind the currently displayed time, to prevent errors related to the internal workings of the time system.

VoIP Subsystem Network Configuration

Parameter	Description	Default
VoIP Startup Delay (ms)	Manually configured VoIP subsystem startup delay. This parameter configures the VoIP subsystem to delay the indicated time before booting up. Normally there is no need to set it. If the VoIP subsystem has trouble registering at power up, you might set this delay to allow your X6v sufficient time to establish a DSL connection before the VoIP subsystem attempts to register.	0
VoIP Name	Manually configured VoIP subsystem device name.	ZOOM_VoIP
VoIP Host Name	Manually configured host device name (or name automatically assigned and saved).	ZOOM_VoIP
VoIP Domain Name	Manually configured domain name.	
MTU	Manually configured maximum transmit unit size (range of 576 to 1500). **Note**: the MTU setting is prepared for the use of PPPoE. Some system configurations require an MTU setting of 1500.	1492

Static IP/DNS Configuration

Parameter	Description	Default
Static IP Address	Manually configured IP address (or address automatically assigned and saved)	192.168.0.234
Subnet Mask	Manually configured local network mask (or netmask automatically assigned and saved)	255.255.255.0
Gateway IP Address	Manually configured gateway IP address (or address automatically assigned and saved)	192.168.0.1
Primary DNS Address	Manually configured IP address of primary domain name server (DNS)	192.168.0.1

Note: You must change the VoIP Subsystem IP/DNS configuration settings first to the new subnet if you change the LAN subnet of your X6v.

HTTP / Telnet / FTP Server

Parameter	Description	Default
HTTP Server Access Enable	Enable or disable access to Configuration Manager	Enabled
HTTP Server Port	Assigned port number for HTTP server	8080
Telnet Server Enable	Enable or disable remote access via telnet	Enabled
Telnet Server Port	Assigned port number for Telnet server	8023
FTP Server Enable	Enable or disable remove access via ftp	Enabled
FTP Server Port	Assigned port number for FTP server	8021

Note: External access may be blocked by your X6v firewall.

STUN Settings

Parameter	Description	Default
STUN Enable	Enables or disables use of STUN for discovery of **N**etwork **A**ddress **T**ranslation (**NAT**) mapping	Enabled
STUN Server Address	Fully qualified domain name (including optional port number) for the STUN server	

Firmware and Configuration Update Settings

Parameter	Description	Default
Update Server Domain Name	Fully qualified domain name (including an optional port number) for the update server	zoom.voipconfigure.com: 5080
Automatic Configuration Update Enable	Control to enable automatic updating of configuration	Enabled
Automatic Configuration Update on Reboot	Control to enable automatic update of configuration on reset	Enabled
Automatic Configuration Update (SIP)	Control to enable automatic update on receipt of SIP message	Disabled
Configuration Update Message on Request	SYSLOG message body sent when requesting a configuration update	Configuration update requested
Configuration Update Message on Success	SYSLOG message body sent when configuration update completed successfully	Configuration update successful
Configuration Update Message on Failure	SYSLOG message body sent when configuration update completed unsuccessfully	Configuration update failed

Parameter	Description	Default
Configuration Update Periodic Delay(s)	Periodic delay between configuration update checks (in seconds - limit 4,294,967,296)	76400
Configuration Update Random Delay(s)	Uniform random delay applied when contact with the update server fails	240
Configuration Update Error Retry Delay(s)	Fixed delay applied when the configuration update operation fails	120
Automatic Firmware Update Enable	Control to enable automatic updating of firmware	Enabled
Automatic Firmware Update on Reboot	Control to enable automatic update of firmware on reset	Enabled
Firmware Update Message on Request	SYSLOG message body sent when requesting a firmware update	
Firmware Update Message on Success	SYSLOG message body sent when firmware update completed successfully	Firmware update successful
Firmware Update Message on Failure	SYSLOG message body sent when firmware update completed unsuccessfully	Firmware update failed
Firmware Update Periodic Delay(s)	Periodic delay between firmware update checks (in seconds - limit 4,294,967,296)	86400
Firmware Update Random Delay(s)	Uniform random delay applied when contact with the update server fails (in seconds)	240
Firmware Update Error Retry Delay(s)	Fixed delay applied when the firmware update operation fails (in seconds)	120

Note: The configuration and/or firmware update periodic delay is by default about a day. This can be changed to a week by specifying 604,800 seconds, or a month by specifying 2,620,800 seconds.

VoIP System Maintenance

Parameter	Description	Default
Syslog Enable	Enable or disable transmission of SYSLOG messages	Disabled
Syslog Server Address	Fully qualified domain name (including an optional port number) for the SYSLOG server	
Debug Enable	Enable or disable transmission of Debug messages	Disabled
Debug Server Address	Fully qualified domain name (including an optional port number) for the Debug server	
Debug Level ATA	VoIP Subsystem debug	0

Parameter	Description	Default
Debug Level SIP	S**ession I****nitiation** P**rotocol debug**	**0**
Debug Level Net	Network debug	0
Debug Level PMP	**P**ort **M**apping **P**rotocol debug	0

4
VoIP Accounts

You can use the **VoIP > Advanced VoIP Setup > VoIP Accounts** menu to configure user accounts for up to four providers. The menu items include:

- My VoIP Account
- Accounts 2, 3, and 4

Logging in to the Configuration Manager

To access the **VoIP** menus, you must log in to the **Configuration Manager**.

1 Type **http://192.168.0.1** in your browser's address field.

2 When prompted, log on in administrator mode, using the following Username and Password:

Username:	**admin**
Password:	**zoomadsl**

3 When the **ADSL Setup** page opens, click the **VoIP** icon on the Zoom menu bar to access the VoIP Subsystem.

4 Click the **Advanced VoIP Setup** icon, then select **VoIP Accounts** to view or modify parameters.

Notes to service providers:

If you are going to lock units to your service, we strongly recommend that you change the admin password before shipping any product to the field. See **Changing the admin Password** on page 6 for instructions.

As an added precaution, we recommend that you also change the VoIP subsystem password. Please refer to the deployment package for details.

Setting User Privileges

You may set access to account information for the user level login (see Setting User Access Privileges, on page 7). For example, you may wish to hide (privilege -) or to make read-only (privilege V) access to My VoIP Account, and allow full access (privilege E) to accounts 2, 3 and 4. Alternatively, you may want to hide access to all four accounts.

On each of the account pages there is a column of priority settings on the right-hand side. The top setting determines access for that page as a whole. The remaining settings determine the privileges of the individual parameters that they control.

There is a limitation in the implementation of the privileges of the individual parameters. These must all be the same for all four accounts. Thus, you should set the individual parameters to support the level of access you wish to grant for the account(s) with the most open access. You may restrict the access to other accounts by choosing an appropriate value for the top level setting that controls those pages.

The **VoIP Express Setup** page is affected by settings on the **My VoIP Account** page. Six parameters on this page are drawn from the **My VoIP Account** page, Turn **My VoIP Service** (On/Off), and the five parameters beginning **My ...**. The user is granted the same access to these parameters through the **Express** page as through the **My VoIP Account** page. (The **VoIP Express Setup** page offers control or view of a subset of settings that are appropriate for many users).

Note: In some fields you might see default values that were used in Zoom's manufacturing test procedures. You can safely ignore or delete these values.

My VoIP Account

Parameter	Description	Default
Turn My VoIP Service	Enables (**On**) or disables (**Off**) this account	**On**
My VoIP Providers Name	Name of VoIP provider	
My Caller ID When I Call Someone	Holds an identifier (name or number) that can be displayed at the receiving party's phone when someone makes a call from the VoIP Subsystem to another SIP phone. When someone makes a call from the VoIP Subsystem that terminates on the PSTN, this ID will generally not display on the receiving party's phone.	
My VoIP Phone Number (SIP User ID)	Specifies the name to be used when logging in to the service provider's server. Commonly implemented in the form of an E.164 number. (E.164 is the ITU recommendation for standard telephone number format.) This ID/number will often appear on the receiving party's phone as the Caller ID when someone places a call from the VoIP Subsystem.	
My VoIP Service Authorization ID	User name for authentication	
My VoIP Service Authorization Password	User password for authentication	
SIP Server	Identifies the SIP Server (Format: FQDN)	

Parameter	Description	Default
Auth Domain	Specifies the authentication domain name corresponding to the Authentication User Name. This field must match the authentication realm URL assigned by the service provider. It must NOT be translated into any dotted-decimal address equivalent. For many service providers, this can be left blank as SIP messages in the registration process will convey the authentication domain name. (Format: FQDN)	
Outbound Proxy	Identifies the outbound proxy server and port, or if the provider doesn't use an outbound proxy server, the default SIP proxy server and port to be used when making outgoing calls. (Format: FQDN)	
Register Domain	Identifies the default SIP registration server name and port used to identify the VoIP Subsystem device providing the service end-point for the assigned subscription service. (Format: FQDN)	
ReReg Interval (s)	Sets the default registration update period in seconds. The VoIP Subsystem must re-register before this period expires to prevent service interruption.	120
Subscribe Domain	Fully qualified domain name (with optional port number) for the SIP registration server. (Format: FQDN)	
ReSub Interval (s)	Re-subscription interval in seconds	1800
Use Outbound Proxy for REFER	Enables or disables the use of an outbound proxy for SIP service remote call transfers	Disabled
DNS Server Lookup for SIP Server	Enables or disables DNS Server lookup services for the SIP server	Disabled
Ring Type	Selects a distinctive ring type for the account.	1

Parameter	Description	Default
Dial Prefix	Contains the dial string pattern matching used to distinguish and route calls to a VoIP service provider. The default for **My VoIP Account** is null (that is, all calls are routed via this account, unless preceded by a prefix defined for accounts two through four). Accounts 2 through 4 can be configured with prefixes that are used to invoke these accounts. The dial string pattern match is in the standard form. Prefix strings of #8, #9, 8 and 9, if specified, are automatically removed from the dialed number. Other prefixes can be altered through the substitution flexibilities of the pattern matching strings.	
Preferred Codecs	Allows listing, in order of preference, the Codec code points preferred for use with the service provider. Menu options are: **G.711u**, **G.711A**, **G.729B**, and **iLBC**. The codecs listed here **must** also be included in the **Preferred Codecs** list under **Audio Settings** on the **VoIP Parameters** page. If any codecs are listed here, then only those codecs will be negotiated. If no codecs are listed here, then all **Preferred Codecs** options will be negotiated.	

Accounts 2, 3, and 4

Parameter	Description	Default
Turn My VoIP Service	Enables or disables this account	Disabled
My VoIP Provider Name	Name of VoIP provider	

Parameter	Description	Default
My Caller ID When I Call Someone	Holds the number that can be displayed at the receiving party's phone when the user makes a call from the VoIP Subsystem to another SIP phone. When the user makes a call from the VoIP Subsystem that terminates on the PSTN, this name will generally not display on the receiving party's phone.	
My VoIP Phone Number (SIP User ID)	Specifies the name to be used when logging in to the service provider's server. Commonly implemented in the form of an E.164 number. This ID/number will often appear on the receiving party's phone as the Caller ID when someone places a call from the VoIP Subsystem.	
My VoIP Service Authorization ID	User name for authentication	
My VoIP Service Authorization Password	User password for authentication	
SIP Server	Identifies the SIP Server. (Format: FQDN)	
Auth Domain	Specifies the authentication domain name corresponding to the user's **Authorization ID**. This field must match the authentication realm URL assigned by the service provider. It must NOT be translated into any dotted-decimal address equivalent. For many service providers, this can be left blank as SIP messages in the registration process will convey the authentication domain name. (Format: FQDN)	
Outbound Proxy	Identifies the outbound proxy server and port, or if the provider doesn't use an outbound proxy server, the default SIP proxy server and port to be used when making outgoing calls. (Format: FQDN)	
Register Domain	Identifies the default SIP registration server name and port used to identify the VoIP Subsystem device providing the service end-point for the assigned subscription service. (Format: FQDN)	

Parameter	Description	Default
ReReg Interval (s)	Sets the default registration update period in seconds. Once the period has expired, the VoIP Subsystem must re-register to prevent service interruption.	120
Subscribe Domain	Fully qualified domain name (with optional port number) for the SIP registration server. (Format: FQDN)	
ReSub Interval (s)	Re-subscription interval in seconds	120
Ring Type	Selects a distinctive ring type for the account.	2 for Account 2 3 for Account 3 4 for Account 4
Dial Prefix	Contains the dial string pattern matching used to distinguish and route calls to a VoIP service provider. The default is null (i.e., all calls are routed via this account, unless preceded by a prefix defined for accounts two through four). Accounts 2, 3, and 4 can be configured with prefixes that are used to invoke these accounts. The dial string pattern match is in the standard form. Prefix strings of #8, #9, 8 and 9, if specified, are automatically removed from the dialed number. Other prefixes can be altered through the substitution flexibilities of the pattern matching strings.	
Preferred Codecs	Allows listing, in order of preference, the Codec code points preferred for use with the service provider. Menu options are: **G.711u**, **G.711A**, **G.729B**, and **iLBC**. The codecs listed here **must** also be included in the Preferred Codecs list under **Audio Settings** on the **VoIP Parameters** page. If any codecs are listed here, then only those codecs will be negotiated. If no codecs are listed here, then all **Preferred Codecs** options will be negotiated.	

5
VoIP Parameters

You can use the **VoIP > Advanced VoIP Setup > VoIP Parameters** menu to configure various common aspects of the VoIP Subsystem device. The menu items include:

- Audio Settings
- RTP Protocol Parameters
- SDP Protocol Parameters
- SDP Audio Codec Names

Audio Settings

Parameter	Description	Default
Preferred Codecs	Lets you arrange the Codec names in order of preference. These entries must agree with the Preferred Codecs specified on the **My VoIP Account** page.	G.711u, iLBC, G.729B, G.711A
Silence Suppression Enable	Prevents audio frames from being sent during periods of silence, thus reducing the network traffic necessary for making calls. (Note: This feature is useful only with audio codecs that support silence suppression.)	Disabled
Echo Canceller Enable	If enabled, the G.168 echo canceller is applied to all calls.	Enabled
Echo Canceller Mode	Sets the echo canceller operating mode.	Do not change the setting, which is 2.
Echo Canceller Tail Length (ms)	Specifies length of echo canceller in msec	16
Fax Transmission Mode	Control for FAX processing method: Off, or Passthrough (µLaw or ALaw)	Off
DTMF Transmission Method	Control for DTMF processing method: Off, Audio Passthrough, RTP Out-of-band, SIP Out-of-band	RTP Out-of-band
iLBC High Rate Enable	Enables 15.2 kbps / 20 ms frames. When disabled, 13.33 kbps / 30 ms frames. Many implementations negotiate 13.33 kbps / 30 ms only.	Disabled

RTP Protocol Parameters

Parameter	Description	Default
Base RTP port (1024-65535)	The minimum IP port number for RTP traffic. Can be used in conjunction with firewall mappings.	1234
Maximum RTP port (1024-65535)	The maximum IP port number for RTP traffic.	1253
RTP Public External IP Address	Forces a specific external IP address as the source address for SDP messages that the VoIP Subsystem sends.	0.0.0.0
RTP Public External Port	Specifies the RTP port associated with the minimum RTP port number in a NAT firewall that performs fixed port mapping.	0 (Disabled)
RTP TOS Value (0x00-0xff)	Type of service (TOS) value or DIFFServ DSCP used for RTP (audio) packets.	68 (Assured Forwarding)
RTP Packet Duration (ms)	The duration (in milliseconds) for frame-based codecs	30
RTP Stream Duration (ms)	The duration (in milliseconds) for sample stream-based codecs	20
RTP Session Timeout Interval (s)	The session timeout interval (in seconds)	120
RTP Jitter Buffer Start Depth (ms)	The start depth (in milliseconds) of the buffer	20
RTP Jitter Buffer Minimum Depth (ms)	The minimum depth (in milliseconds) of the buffer	20

SDP Protocol Parameters

Parameter	Description	Default
SDP Session Name	Identifies the session name.	-
SDP Session Owner	Identifies the session owner.	Zoom

SDP Audio Codec Names

These parameters are passed to the remote end-point for outgoing calls only.

Parameter	Description	Default
G711u Codec (PCMU/8000)	The string passed during outgoing calls to negotiate the payload type for G.711 µLaw	PCMU/8000
G711A Codec (PCMA/8000)	The string passed during outgoing calls to negotiate the payload type for G.711 ALaw	PCMA/8000
G729b Codec (G729B/8000)	The string passed during outgoing calls to negotiate the payload type for G.729B	G729B/8000
iLBC/Codec (iLBC/8000)	The string passed during outgoing calls to negotiate the payload type for iLBC	iLBC/8000

6
SIP Parameters

You can use the **VoIP > Advanced VoIP Setup > SIP Parameters** menu to configure particular aspects of the **S**ession **I**nitiation **P**rotocol (**SIP**) implementation. The menu items include:

- SIP Protocol Parameters
- SIP Response Codes
- SIP Distinctive Ring Names
- SIP Protocol Timers

SIP Protocol Parameters

Parameter	Description	Default
SIP Require User Name	Enables or disables a requirement that an incoming INVITE include a SIP user name assigned to the VoIP subsystem in an active account.	Disabled
SIP Local Port (1024-65535)	Local UDP port used for sending/ receiving SIP call control messages. This port can be mapped by a firewall.	5060
SIP Public External IP Address	Forces a specific external IP address for SIP messages sent	0.0.0.0 (Disabled)
SIP Public External Port	Forces a specific external UDP port for SIP messages sent	0 (Disabled)
TOS Value (0x00 – 0xff)	Type of service (TOS) value or DIFFServ DSFIELD used for SIP message	68 (DIFFSRV Expedited Forwarding)
SIP Accept Language String	Specifies the language for user-viewable messages used in the SIP accept message	English
SIP Send Response to SRC Port	Respond to the sender’s IP address/UDP port used by SIP request message	Enabled
SIP Max Forwards	Maximum forward value	15
SIP Ringing Retransmit	Enables or disables retransmission	Enabled
SIP Use NAT Discovery	Enable use of NAT discovery procedures to obtain an external IP address/UDP port mapping for SIP messages	Enabled
SIP Use Received Via Info	Use VIA header IP address/UDP port parameters in received messages as external IP address/UDP port	Disabled

Parameter	Description	Default
NAT Keep Alive Enable	Send periodic SIP messages to keep port mapping active	Enabled
NAT Keep Alive Interval (s)	Periodic interval for SIP keep alive messages (in seconds)	15
NAT Keep Alive Domain Name	Fully qualified domain name (including an optional port number) for the destination of SIP keep alive message (sends to the proxy server if blank)	
NAT Keep Alive Message	Type of message to be sent as SIP keep alive: empty, notify or register	

SIP Response Codes

Parameter	Description	Default
SIP Response Code SIT1	SIP response code which plays the SIT1 tone sequence	0
SIP Response Code SIT2	SIP response code which plays the SIT2 tone sequence	0
SIP Response Code SIT3	SIP response code which plays the SIT3 tone sequence	0
SIP Response Code SIT4	SIP response code which plays the SIT4 tone sequence	0
SIP Response Code Try Backup	SIP response code to use backup server	0
SIP Response Code Retry Registration	SIP response code to retry the registration	30

Note: The range for the **SIP Response Codes** is 0 through 65535. However, the **SIP Response Codes** are not implemented.

SIP Distinctive Ring Names

Parameter	Description	Default
01	Telephone event name to produce distinctive ring pattern 1	Belcore-r1
02	Telephone event name to produce distinctive ring pattern 2	Belcore-r2
03	Telephone event name to produce distinctive ring pattern 3	Belcore-r3
04	Telephone event name to produce distinctive ring pattern 4	Belcore-r4
05	Telephone event name to produce distinctive ring pattern 5	Belcore-r5
06	Telephone event name to produce distinctive ring pattern 6	Belcore-r6
07	Telephone event name to produce distinctive ring pattern 7	Belcore-r7
08	Telephone event name to produce distinctive ring pattern 8	Belcore-r8

SIP Protocol Timers

Parameter	Description	Default
SIP Timer INVITE Expires (s)	The time (in seconds) after which an INVITE request expires.	180
SIP Timer Re -INVITE Expires (s)	The time (in seconds) after which a retransmitted INVITE request expires.	180
SIP Timer Registration Min (s)	The minimum Registration Period (in seconds).	1
SIP Timer Registration Max (s)	The maximum Registration Period (in seconds).	7200
SIP Timer Registration Retry (s)	The time interval (in seconds) for retrying a (failed) REGISTER request.	30
SIP Timer No Answer Duration (s)	The length of time (in seconds) before terminating a session request.	60
SIP Timer Re-Register Interval (s)	The elapsed time (in seconds) between an initial and repeat REGISTER request.	20
SIP Session Timer (s)	The time interval (in seconds) for the session timer.	0

Note: The range for the **SIP Protocol Timers** is 0 through 65535. However, the **SIP Protocol Timers** are not implemented.

7 Regionalization

You can use the **VoIP > Advanced VoIP Setup > Regionalization** menu to configure the VoIP Subsystem for local operating conventions. The menu options include:

- Call Progress Tones
- Standard Ringing Patterns
- Distinctive Ringing Patterns
- Distinctive Call Waiting Patterns
- Voice and Tone Parameters
- SLAC Configuration
- SLAC Command Strings
- CODEC Configuration
- CODEC Command Strings
- Other

Note: In some fields below you might see default values that are valid for the United States only. If you are reviewing or configuring VoIP settings for other regions, those default values do not apply.

Call Progress Tones

Call progress tones are specified by a list of values indicating the number of tones, number of on/off transitions, frequency/signal level pairs, and tone on/off times. The format is:

> ***no_of_tones, no_of_times, duration,***
> ***{tone_element1_freq, tone_element1_db, tone_element2_freq, tone_element2_db, …},***
> ***{tone_on_time1, tone_off_time1, tone_on_time2, …}***

where:

no_of_tones is the number of tone elements that are combined to form a tone. Each tone element has an associated frequency and amplitude. Up to four tone elements can be combined – to form a chord, or played in sequence – as a tune (see ***no_of_times***). A negative ***no_of_tones*** indicates that the tones will be synchronized to a two-second timer (relevant for multi-port ATAs only).

no_of_times is the total of both on-to-off and off-to-on transitions in the tone pattern. If this value is positive, it produces a composite tone. If it is negative, the tones are played in sequence. Zero produces a continuous composite tone

duration is the length of time in seconds that the call progress tone will be played. A value of zero means that the tone will be played until instructed otherwise.

tone_elementX_freq and ***tone_elementX_db*** represent the frequency (Hz) and signal level (dB) of each tone. A negative frequency is used to modulate the prior tone components summed together.

A negative dBm level can be offset by *ipbx_tone_gain*. Allowed values for ***freq*** are from 0 to 3000Hz. Allowed values for ***db*** levels are from –1 to –40 dB.

tone_on_timeX and ***tone_off_timeX*** are interleaved Tone On and Tone Off durations in msec. A value of zero for a Tone On time indicates a continuous tone. A value of zero for a Tone Off time produces silence, while a negative value (-1) terminates the tone pattern, removing the silencing. (With silencing, the voice channel is blocked until the tone pattern is stopped.) The maximum number of tones is four. The maximum number of on-to-off and off-to-on times counted individually is nine.

For example, the default setting for initial North American dial tone is:

{2, 0, 0, {350, -19, 440, -19}, {0}}

where:

2 is the number of frequency/dB pairs (**350, -19,** and **440, -19**)

The first **0** is the number of on/off transitions in the tone pattern, in this case a constant tone.

The second **0** indicates that the tone will be played until otherwise instructed.

The first pair of frequency/dB (**350, -19**) specifies that the first tone is at 350Hz with a level of -19dB.

The second pair of frequency/dB (**440, -19**) specifies that the second tone is at 440Hz with a level of -19dB.

The final **{0}** specifies that there are no on/off times and that the tone is constant.

Call Progress Tone Parameters

Parameter	Description	Default (North America)
Initial Dial Tone	The default tone used when a person begins any dialing operation	2 0 0 350 -19 440 -19
Alternate Dial Tone	The alternate tone used when a person begins any dialing operation	1 0 0 400 -16
Secondary Dial Tone	The tone used in cases where a person can dial a number to access a designated type of line	2 0 0 420 -19 520 -19
Stutter Dial Tone	Indicates a message waiting	2 7 0 350 -19 440 -19 100 110 100 110 100 110 0
Message Waiting Dial Tone	Indicates a message waiting	2 2 0 350 -19 440 -19 160 160
Call Forward Dial Tone	Indicates that calls are being forwarded	2 3 0 350 -19 440 -19 250 400 0
Pre-Ringback Tone	Played while a call is being signaled before a confirmation is received from the SIP server	0 0 0 (Silence)
Ringback Tone	Played while a call is connecting	2 2 0 440 -19 480 -19 2000 4000
Call Waiting Tone Default	Played when an incoming call arrives and the phone is in use	1 2 0 440 -16 300 9700
PSTN Call Waiting Tone Default	Played when a call is on hold longer than the timeout hold duration	1 2 0 440 -16 300 9700

Parameter	Description	Default (North America)
Station Call Waiting Tone Default	Call waiting pattern for station to station calls. Applies to multi-port units only.	1 2 0 440 -16 300 9700
Call Holding Tone	Reminder tone that a call is on hold	1 4 0 1200 -16 100 200 100 -1
Call Disconnect Tone	Played when a call on hold has disconnected	1 4 0 350 -16 50 100 50 -1
Call Conference Tone	Played when a conference is in progress	1 2 0 350 -16 100 15000
Busy Tone	Sent back to the caller when the recipient's line is busy	2 2 0 480 -19 620 -19 500 500
Reorder Tone	A fast, busy, or congestion tone sent to the caller when a call cannot go through	2 2 0 480 -19 620 -19 250 250
Off Hook Warning Tone	Sounds when the telephone is off-hook for longer than the timeout alert duration	4 2 0 1400 11 2050 11 2450 11 2600 11 100 100
SIT1 Tone	Sent to the user when a telephone number is invalid or has been disconnected	3 -6 0 985 -16 1428 -16 1777 -16 330 5 330 5 330 1000
SIT2 Tone	Sent to the user when a telephone number is invalid or has been disconnected	3 -6 0 914 -16 1371 -16 1777 -16 330 5 330 5 330 1000
SIT3 Tone	Sent to the user when a telephone number is invalid or has been disconnected	3 -6 0 985 -16 1428 -16 1777 -16 380 5 380 5 380 1000
SIT4 Tone	Sent to the user when a telephone number is invalid or has been disconnected	3 -6 0 914 -16 1371 -16 1777 -16 380 5 380 5 380 1000
Prompt Tone	Played when the user has completed a segment of input	2 0 0 520 -19 620 -19
Confirm Tone	Played when the user has entered an acceptable value	1 2 0 600 -16 400 0
Input Error Tone	Played when the user has made an invalid entry	2 2 0 480 -19 620 -19 250 250
Number Error Tone	Played when the user has entered an invalid dial string	2 2 0 480 -19 620 -19 250 250

Standard Ringing Patterns

Ring patterns are specified by a list of values indicating the frequency, number of on/off transitions, and Ring On/Ring Off times. The format is:

ring_frequency, no_of_times, duration,
{ring_on_time1, ring_off_time1, ring_on_time2, ring_off_time2, …}

where:

ring_frequency specifies the frequency of the ringing tone in Hz for sinusoidal and trapezoidal ringing. This value is only used if the default ringer parameter *slac_ring_frequency* is zero.

no_of_times is the total of both on and off transitions in the ring pattern. This can be zero for a continuous ring signal (which may not be desirable and may exceed the rated power capacity of the ATA).

duration is the length of time in seconds to ring. A value of zero means until instructed otherwise.

ring_on_timeX and ***ring_off_timeX*** are interleaved Ring On and Ring Off durations in msec. A value of zero for a Ring On time indicates a continuous tone. A value of zero for a Ring Off time produces continuous silence.

Possible values for frequency are between 0Hz and 60Hz. The maximum total of on and off times summed together is nine.

Standard Ringing Patterns Parameters

Parameter	Description	Default (North America)
Ring Default	Default ring pattern	20 2 0 2000 4000
PSTN Ring Default	Default PSTN call ring pattern	20 2 0 2000 4000
Station Ring Default	Default station call ring pattern	20 2 0 1000 3000
Call Hold Re-Ring	Call on hold reminder re-ring pattern	20 2 0 500 0
Call Back Ring	Call back success ring pattern	20 2 0 1500 0
Call Back Ring Splash	Call back in progress ring pattern	20 2 0 700 0
Call Forward Ring Splash	Call forward reminder ring pattern	20 2 0 500 0
Message Waiting Ring Splash	Audible message waiting ring pattern. This parameter is for analog telephone adapter products only.	20 2 0 500 0

Distinctive Ringing Patterns

The distinctive ring feature allows different ring patterns to be sent to the telephone according to **Distinctive Ring** parameters 1 - 8. Distinctive ringing patterns are specified in the same way as standard ringing patterns.

The user can assign distinctive ringing patterns to particular callers under User Configuration…Ringing Based on Caller ID.

Distinctive Ringing Patterns Parameters

Parameter	Description	Default (All Regions)
Distinctive Ring 1	Specifies the pattern for Ring 1	20 2 0 2000 4000
Distinctive Ring 2	Specifies the pattern for Ring 2	20 4 0 1000 1000 1000 3000
Distinctive Ring 3	Specifies the pattern for Ring 3	20 6 0 300 200 1000 200 300 4000
Distinctive Ring 4	Specifies the pattern for Ring 4	20 4 0 800 400 800 4000
Distinctive Ring 5	Specifies the pattern for Ring 5	20 4 0 400 200 400 2000
Distinctive Ring 6	Specifies the pattern for Ring 6	20 2 0 1000 3000
Distinctive Ring 7	Specifies the pattern for Ring 7	20 4 0 300 200 1500 2000
Distinctive Ring 8	Specifies the pattern for Ring 8	20 4 0 800 400 800 2000

Distinctive Call Waiting Patterns

A call waiting tone is played when an incoming call arrives while the phone is in use. Support for up to eight distinctive call waiting tone patterns is available. Distinctive call waiting patterns are specified in the same way as standard ringing patterns.

When the user assigns a distinctive ringing pattern to a particular Caller ID, the corresponding distinctive call waiting pattern is also assigned to that Caller ID.

Distinctive Call Waiting Patterns Parameters

Parameter	Description	Default (North America)
Call Waiting Tone 1	Specifies the pattern for Tone 1	1 2 0 440 -16 300 9700
Call Waiting Tone 2	Specifies the pattern for Tone 2	1 6 0 440 -16 100 20 100 20 100 9660
Call Waiting Tone 3	Specifies the pattern for Tone 3	1 4 0 440 -16 100 100 100 9700
Call Waiting Tone 4	Specifies the pattern for Tone 4	1 6 0 440 -16 100 100 100 100 100 9500
Call Waiting Tone 5	Specifies the pattern for Tone 5	1 2 0 620 -16 300 9700
Call Waiting Tone 6	Specifies the pattern for Tone 6	1 6 0 620 -16 100 20 100 20 100 9660
Call Waiting Tone 7	Specifies the pattern for Tone 7	1 4 0 620 -16 100 100 100 9700
Call Waiting Tone 8	Specifies the pattern for Tone 8	1 6 0 620 -16 100 100 100 100 100 9500

Voice and Tone Parameters

The parameters in the following sections control the connection to the local phone (FXS) port on the VoIP Subsystem. This includes control of both the Subscriber Line Interface Circuit (SLIC) and **S**ubscriber **L**ine **A**udio **C**ircuit (**SLAC**) that together make up the FXS port.

Voice, Tone and DTMF Parameters

The following table lists parameters that control voice and tone signals, transmit and receive levels, and **D**ual-**T**one **M**ulti-**F**requency (**DTMF**) signaling tone characteristics.

Parameter	Description	Default
Voice RX Gain (-20 to +10 dB)	Voice receive gain in dB	0
Voice TX Gain (-20 to +10 dB)	Voice transmit gain in dB	0
Tone Gain (-20 to +10 dB)	Tone signal gain in dB (applied to locally generated tones such as call paging tones).	0
Tone Max (-20 to 0 dBm)	When two tones of equal amplitude are added together, the signal level is 3dB higher than the individual components. When four tones of equal amplitude are added together, the signal level is 6dB higher than the individual components. This limit prevents inadvertent saturation and user hearing damage.	-12
DTMF Low Tone Gain (-20 to -5 dBm)	Low frequency group DTMF tone level in dBm	-9
DTMF High Tone Gain (-20 to -5 dBm)	High frequency group DTMF tone level in dBm	-7
DTMF Tone On Time (ms)	DTMF generation On time (50 to 200 ms)	80
DTMF Tone Off Time (ms)	DTMF generation Off time (50 to 200 ms)	50
DTMF Detect ABCD	DTMF detection enable for ABCD dual tone pairs	Enabled
DTMF Generate ABCD	DTMF generation enable for ABCD dual tone pairs	Enabled
DTMF Pad Duration (ms)	DTMF out-of-band On time in milliseconds (0 to 10,000 ms)	100
DTMF Wait Duration (ms)	DTMF out-of-band Off time in milliseconds (0 to 10,000 ms)	50
DTMF Playout Min Duration (ms)	DTMF out-of-band minimum on time in milliseconds (0 to 10,000 ms)	100

SLAC Configuration Parameters

Parameter	Description	Default
Port Impedance (See the **Values for Port Impedance** table.)	Synthetic impedance matching network control for a choice of one of 10 common world-wide configurations	Varies by region
Port RX Gain (GR) (-12 to 6dB)	SLAC receive gain in dB units	-1
Port TX Gain (GX) (+12 to 6dB)	SLAC transmit gain in dB units (**Note**: A value of 6 dB of attenuation is automatically applied by the GX gain block prior to the specified transmit gain.)	5
Audio Clamp Duration (ms)	Audio clamp On time (0 to 65535 ms)	100
Caller ID Type 1 Mode	Caller ID type 1 (on-hook) mode (None, Belcore MDMF, Belcore SDMF, ETSI Wink, ETSI Ring, DTMF)	Belcore MDMF
Caller ID Type 2 Mode	Caller ID type 2 (off-hook) mode (None, Belcore MDMF, Belcore SDMF, ETSI Wink, ETSI Ring, DTMF)	Belcore MDMF
Message Waiting Mode	Message waiting mode (None, Belcore MDMF Belcore SDMF, ETSI)	Belcore MDMF
Ring Type	Selects ring waveform type: Sinusoidal or Trapezoidal	Sinusoidal
Ring Frequency (0 to 60 Hz)	Ringer frequency in Hz (zero to use ring pattern frequency specification)	0
Ring Transition (ms)	Trapezoidal transition time (0 to 1000ms)	15ms
Ring Amplitude (v)	Ringer voltage in volts (-155v to +1.55v)	85v
Ring Bias (v)	Ringer bias in volts DC (-155v to +1.55v)	0
Message Waiting Type	Selects visual message waiting waveform type: Sinusoidal or Trapezoidal	Sinusoidal
Message Waiting Frequency (Hz)	Visual message waiting frequency in Hz (0 to 60Hz)	25Hz
Message Waiting Transition (ms)	Trapezoidal transition time in msec (0 to 1000ms)	15ms
Message Waiting Amplitude (v)	Visual message waiting voltage in volts (-155v to +155.v)	50v
Message Waiting Bias (v)	Visual message waiting bias in volts (-155v to +155v)	0

Values for Port Impedance (SLAC & CODEC)

Index	Impedance	Country
0	**600 (default)**	US
1	**900**	US/Canada
2	**600 + 1.0 μF**	
3	**900 + 2.16 μF**	

Index	Default	Country
4	**270 + 750 || 150 nF**	Sweden/CTR21
5	**220 + 820 || 120 nF**	Germany/Austria/Australia/New Zealand #1
6	**220 + 820 || 115 nF**	Bulgaria/South Africa/Slovakia
7	**370 + 620 || 310 nF**	UK #1/India/New Zealand #2
8	**200 + 680 || 100 nF**	China
9	**800 || 50 nF**	

SLAC Command Strings

The SLAC initialization commands provide a method to set up the device for unusual conditions. Do not change the default value unless the factory has suggested you do so.

Parameter	Description	Default
Initialization Commands	Specifies device setup for unusual conditions.	100

CODEC Configuration

This section describes the TELCO (FXO) port connection on the VoIP Subsystem and the CODEC (COde DECode) configuration that provides the signal interface to the FXO port.

Parameter	Description	Default
Port Impedance (See the **Values for Port Impedance** table on page 30)	Synthetic impedance matching network control for a choice of one in ten common world-wide configurations	Default
Port RX Gain (GR) (-12 to +6 dB)	SLAC receive gain in dB units	0
Port TX Gain (GX) (-12 to +12 dB)	SLAC transmit gain in dB units (**Note**: 6dB of attenuation is automatically applied by the GX gain block prior to the specified transmit gain.)	-2
Audio Clamp Duration (ms)	Audio clamp On time in milliseconds (0 to 65535ms)	300
Line in Use Detect Method	Defines the method to use for detecting the TELCO line's status.	Default
Line in Use Inhibit	Enables or disables use of the TELCO line.	Disabled
Parallel in Use Debounce	Specifies the number of lines that can be used in parallel. 0 to 65535 lines are the possible min/max values; however, the physical limit is 5.	4
Parallel in Use Detect Method	Defines the method to use for detecting the availability of a parallel line.	Default
Parallel in Use Disconnect	Enables or disables disconnection of a parallel line.	Disabled

Parameter	Description	Default
Caller ID Type 1 Mode	Caller ID type 1 (on-hook) mode (None, Belcore MDMF, Belcore SDMF, ETSI WINK, ETSI RING, DTMF)	Belcore MDMF
Caller ID Type 2 Mode	Caller ID type 2 (off-hook) mode (None, Belcore MDMF, Belcore SDMF, ETSI WINK, ETSI RING, DTMF)	Belcore MDMF
Message Waiting Mode	Message waiting mode (None, Belcore VMWI, ETSI, Low Voltage Ring)	Belcore VMWI
Ring Detect Duration (ms)	The range is 0 to 65535 ms	100 ms
Ring Detect Period Minimum (ms)	The range is 0 to 65535 ms	18 ms
Ring Detect Period Maximum (ms)	The range is 0 to 65535 ms	64 ms
Ring Detect Threshold	The range is 0 to 65535 ms	0
Ring Silence Period	The range is 0 to 10,000 ms	5200 ms
Ring Minimum period (ms)	The range is 0 to 10,000 ms	1500 ms
Disconnect Voltage Enable	Disconnect on on-hook voltage	Enabled
Disconnect Voltage Duration (ms)	The range is 0 to 10,000 ms	100 ms
Disconnect Polarity Enable	Disconnect on TIP/RING reversal	Enabled
Disconnect Reversals Answer	The range is 0 to 10	1
Disconnect Reversals Originate	The range is 0 to 10	2
Disconnect Silence Enable	Interpret silence on line as disconnect	Disabled
Disconnect Silence Duration (s)	The range is 0 to 10,000 s	15 s
Disconnect Silence Threshold	The range is -32768 to +32767 dB m0	-40
Disconnect Tone1 Mode	Select Mode (Dial Tone, Busy, or other)	Dial Tone
Disconnect Tone 1 Definition	Definition as per Call Progress tones	2 0 0 350 - 19 440 - 19
Disconnect Tone 1 Duration (ms)	The range is 0 to 10,000 ms	5000 ms
Disconnect Tone 1 Bandwidth (Hz)	The range is 0 to 100 Hz	30 Hz
Disconnect Tone 2 Mode	The range is 0 to 100 Hz	Busy Tone
Disconnect Tone 2 Definition	The range is 0 to 100 Hz	2 2 0 480 - 19 620 - 19 500 500
Disconnect Tone 2 Duration (ms)	The range is 0 to 10,000 ms	3000 ms
Disconnect Tone 2 Bandwidth (Hz)	The range is 0 to 100 Hz	30 Hz
Disconnect Tone 3 Mode	The range is 0 to 100 Hz	User Defined Tone
Disconnect Tone 3 Definition	The range is 0 to 100 Hz	0 2 0 450 450
Disconnect Tone 3 Duration (ms)	The range is 0 to 10,000 ms	3000 ms
Disconnect Tone 3 Bandwidth (Hz)	The range is 0 to 100 Hz	30 Hz

CODEC Command Strings

The CODEC initialization commands provide a method to set up the device for unusual conditions. Do not change the default value unless the factory has instructed you to do so.

Parameter	Description	Default
Initialization Commands	Specifies device setup for unusual conditions.	100

Other Parameters

Parameter	Description	Default
Hook Debounce (units of 10 ms)	The range is 0 to 65535 ms	10 (that is, 100 ms)
Ring Debounce	The range is 0 to 65535 ms	20 ms
Disconnect Debounce	The range is 0 to 65535 ms	40 ms
Reconnect Debounce	The range is 0 to 65535 ms	20 ms

8
Subscription Services

You can use the **VoIP > Advanced VoIP Setup > Subscription Services** menu to configure the VoIP Subsystem for the specific advanced services permitted and/or supported. The menu items include:

- Subscription Service Settings
- Dialing Parameters
- VoIP and PSTN Dial Patterns

Subscription Service Settings

Parameter	Description	Default
Call Waiting	Enables customer use of call waiting service	Enabled
Caller ID Inbound	Enables customer use of incoming caller ID service	Enabled
Caller ID Outbound	Enables customer use of outgoing caller ID service (i.e. always send caller ID information)	Enabled
Call Waiting Caller ID Service	Enables customer use of incoming caller ID during call waiting service	Enabled
Call Back	Enables customer use of call back service	Enabled
Call Return	Enables customer use of call return service	Enabled
Speed Dial	Enables customer use of speed dial service	Enabled
Do Not Disturb	Enables customer use of do not disturb service	Enabled
Block Anonymous	Enables customer use of anonymous call block service	Enabled
Call Forward Always	Enables customer use of call forward service	Enabled
Call Forward on Busy	Enables customer use of call forward when busy service	Enabled
Call Forward on No Answer	Enables customer use of no answer call forward service	Enabled
Call Forward Priority	Enables customer use of priority call service	Enabled

Parameter	Description	Default
Distinctive Ring	Enables customer use of distinctive ring service	Enabled
Disturb Accept	Enables customer use of do not disturb accept service	Enabled
Blocked Number	Enables customer use of blocked number service	Enabled
Outgoing Block	Enables outgoing blocked number	Enabled
Forward Last Call	Enables customer use of forward to last caller service	Enabled
Distinctive Ring Last Call	Enables customer use of distinctive ring for last caller service	Enabled
Disturb Accept Last Call	Enables customer use of do not disturb accept last caller service	Enabled
Block Last Call	Enables customer use of block last caller service	Enabled
Three-Way Calling	Enables customer use of three way calling service	Enabled
Three-Way Conference	Enables customer use of three way conference service	Enabled
Attended Transfer	Enables customer use of attended call transfer service	Enabled
Unattended Transfer	Enables customer use of unattended call transfer service	Enabled
Message Waiting	If voice mail is enabled, the VoIP Subsystem can send a distinctive dial tone to indicate that there are unplayed messages in the user’s voice mailbox.	Enabled
Visual Message Waiting	Enables customer use of visual message waiting service	Enabled
Remote Feature Code	Enables sending all features codes to remote service provider	Disabled
Default Feature Code	Enables sending all unprocessed feature codes to remote service provider	Disabled

Dialing Parameters

Parameter	Description	Default
Mode	Mode allows selection of treatment of ***** and # as the leading digit of a dial string. These characters may be processed locally, or they may be passed through to the service provider. If there is a requirement that the service provider process commands that start with #, or for sequences such as "*** ***", these characters must be passed through. Select **Normal** for local processing of these digits; **Pass-through** to pass these digits to the service provider. Note that when **Pass-through** mode is selected, feature codes and speed dials cannot be handled locally on the VoIP Subsystem.	**Normal** Interpret ***** and # DTMF tones locally.)
My VoIP Account Unavailable	Standard Dial Tone, Alternate Dial Tone, No Dial Tone	Alternate Dial Tone
No VoIP Accounts Available	Standard Dial Tone, Alternate Dial Tone, No Dial Tone	Alternate Dial Tone
PSTN Not Available	Standard Dial Tone, Alternate Dial Tone, No Dial Tone	No Dial Tone
Dial Direct	Direct dial processing mode (Disallowed, VoIP only, PSTN only, BOTH or DIRECT)	BOTH
Dial After #8	Processing mode after a #8 prefix (Disallowed, VoIP only, PSTN only, BOTH or DIRECT)	DIRECT
Dial after #9	Processing mode after a #9 prefix (Disallowed, VoIP only, PSTN only, BOTH or DIRECT)	VoIP only
Speed Dial	Processing mode for speed dial (Disallowed, VoIP only, PSTN only, BOTH or DIRECT)	VoIP only
VoIP Dial Pattern *(See VoIP and PSTN Dial Patterns on page 37.)*	Pattern match for VoIP dialing	[3469]11|*xx|**|[1-9]e#r5xp3r*x|p8[1-9]e#r5xp3r*x|3[1-9]e#r5xp3r*x|1010Se#p2r*x|0Se#r5xp2r*x
PSTN Dial Pattern *(See VoIP and PSTN Dial Patterns on page 37.)*	Pattern match for PSTN dialing	100|11x|911|999
Configure VoIP Dial Pattern *(See VoIP and PSTN Dial Patterns on page 37.)*	Used to configure how the VoIP Subsystem handles VoIP dial strings.	[3469]11|*xx|**|[1-9]e#r5xp3r*x|p8[1-9]e#r5xp3r*x|3[1-9]e#r5xp3r*x|1010Se#p2r*x|0Se#r5xp2r*x

Parameter	Description	Default
Configure PSTN Dial Pattern *(See VoIP and PSTN Dial Patterns, below.)*	Used to configure how the VoIP Subsystem handles PSTN dial strings.	
Hot Line Dialing	If enabled, the VoIP Subsystem automatically dials the hot/warm dial string as soon as the telephone receiver is picked up.	Disabled
Warm Line Dialing	If enabled, when the telephone receiver is picked up, the VoIP Subsystem automatically dials the hot/warm dial string after a short wait (default is four seconds).	Disabled
Hotwarm Dial String	Used in hot and warm dialing when one or the other is enabled.	
Auto-Add This Area Code ...	Sets the area code to add automatically.	
Polarity Dialing	Sets the SLAC line polarity during dialing (Forward or Reverse).	Forward
Number of Digits I Will Dial For Local Calls	Specifies the default number of digits to be dialed for local calls.	7
Polarity Dialing	Sets the SLAC line polarity during dialing (Forward or Reverse)	Forward
Polarity Dial Tone	Sets the SLAC line polarity during dial tone (Forward or Reverse)	Forward
Polarity Connect	Sets the SLAC line polarity during connect (Forward or Reverse)	Forward
Polarity Answer	Sets the SLAC line polarity during answer (Forward or Reverse)	Forward
Polarity Idle	Sets the SLAC line polarity during idle (Forward or Reverse)	Forward

VoIP and PSTN Dial Patterns

The **VoIP Dial Pattern** and the **PSTN Dial Pattern** together determine how the VoIP Subsystem handles dial strings when someone dials a number from an attached phone. For units without an FXO port, the **PSTN Dial Pattern** is ignored. In a given location, there are generally only a few types of dialed numbers that need to be defined:

- Dialing for local calls
- Dialing for domestic toll calls,
- Dialing for international toll calls.

In addition, there are specific short strings that are set aside for emergency dialing, and there might be other special strings that invoke telephone features.

By default, the VoIP Subsystem is configured to handle number patterns in every country in the world. For models with an FXO port, emergency calls are by default routed to the PSTN, and all other calls are routed via VoIP. If no telephone line is connected to the Telco port, emergency calls are routed via VoIP.

You can use the dial patterns to change which calls are sent via VoIP, and which are sent to the PSTN. For example, you might want to send all local calls via the PSTN, because these might be free on your PSTN line.

You might also want to tailor the dial patterns to precisely reflect the format of telephone numbers in your location. For example, the default configuration recognizes that a local number might be from 5 to 10 digits long. If local numbers are always 8 digits, this means that the VoIP Subsystem will wait a few seconds after the 8^{th} digit has been dialed, to see if any digits follow. You could redefine the local dial string always to expect 8 digits, and to immediately send the number to the service provider once someone had dialed 8 digits.

Dial Pattern Parameters

Parameter	Description
|	Separates patterns.
Any DTMF char or chars	Literal list of one or more DTMF characters to match in the order shown, and in the position indicated within the pattern.
x	Match any numerical digit (0-9)
~	Match any digit (0-9, A-D, *, #) excluding any specified terminators
[]	Selection group of candidate digits. This group can contain any number of DTMF characters, any of which are considered a match.
[^]	Exclusion group of digits. If any DTMF character occurs at this point in the dial string which matches the exclusion digits listed after the carat, the dialed string fails the match test with this pattern.
[0-9]	Selection range of candidate numerical digits
[a-d]	Selection range of candidate letter digits
r	Repeat operator. Syntax **r n p**, where **r** is the repeat operator, **n** is the number of repetitions, and **p** is the item that is repeated. **n** can be 1-9 repetitions, letters a-z for 10 to 35 repetitions or ***** (asterisk)**,** **+** (plus sign) or **.** (period) to mean repeat until the person stops dialing.
.(period)	Repeat the previous digit until the person stops dialing.
+	Repeat the previous digit until the person stops dialing.
!	Disallows pattern. This element can prevent users from dialing numbers or classes of numbers.
$	Indicates secondary dialing to follow - used only by fixed dial strings.
<:>	Replace group: replace digits to the left of the colon with those to the right.
s	Seize on string as only candidate if dialed digits match to this point.
e	Specify ending termination digit which follows (usually ***** or #). When the user dials the ending termination digit, the VoIP Subsystem considers the dial string complete, and immediately sends to the service provider the digits up to the termination character.
f	Pause timeout causes failure instead of dial.
p	Pause Operator. Syntax **p n**, where **n** is the time in seconds to allow between digits dialed. If this time is exceeded, the dialing is considered to have timed out, and the person to have stopped dialing.

Parameter	Description
t	Set digit timeout to default for current pattern.
- (dash)	Human-readable spacing which is ignored.
(space)	Human-readable spacing which is ignored.

Notes:

Interdigit timeout, or pause: By default, the device allows five (5) seconds between dialed digits. To change this default, you must insert the **p** parameter before the point in the match string that you want this parameter to change.

For example, if you would like a nine (9) second delay after each digit is pressed, then you would need to enter **p9** at the beginning of the pattern matching string. Similarly, if you would like a shorter timeout of three (3) seconds towards the end of a dial string, you would need to enter **p3** before the last entry in the pattern matching string: **...p3r*x.**

Examples of Dial Strings

Each parameter in a pattern match string represents a single digit. The only exceptions are parameters that include a repeat operator. We will illustrate these features by examining several entries in the default VoIP dial string:

[346]11|*xx|[1-9]e#r5xp3r*x|p8[1-9]e#r5xp3r*x|#[1-9]e#r5xp3r*x|1010Se#p2r*x|0Se#r5xp2r*x[3469]11**

Entries are separated by the pipe "|" character. Each entry represents a possible match to the digits that someone dials.

The following descriptions explain how some of the entries in the default Dial String behave.

[346]11 indicates to recognize the sequences **311**,**411**, **611** and **911**, and send them to the service provider when complete.

***xx** is a string that allows the VoIP Subsystem to recognize and forward feature codes to the service provider. However, note that by default, feature codes are handled locally, in the VoIP Subsystem. The VoIP Subsystem refers to this string only if the remote or default feature code parameters are enabled, or if **Passthrough** mode is enabled. In those cases, this string must be included in the pattern matching string, so that the VoIP Subsystem will forward feature codes to the service provider.

****[1-9]e#r5xp3r*x** is a string that pertains to VoIP provider area codes. The ****** prefix is a signal for the service provider to forward this call to another VoIP service provider. The three digits following ****** constitute the VoIP provider area code. Recognize a string starting with ******, and proceeding with any of the digits 1-9. **e**# defines # as the terminating character. If someone dials # at any point after the 1-9, the VoIP Subsystem sends out all digits dialed to that point to the service provider. If the person doesn't dial a #**,** collect five more digits (**r5x**), switch from the default inter-digit timeout of five (5) seconds to a shorter inter-digit timeout of three (3) seconds (**p3**), and continue collecting digits until a timeout occurs (**r*x**). This string will be forwarded only if the VoIP Subsystem is in **Passthrough** mode.

p8[1-9]e#r5xp3r*x is the workhorse string of the default pattern for dialing. It matches dialing for VoIP calls, and for local dialing in most countries. It also matches dialing for domestic long distance dialing under the North American dial plan. This string is identical to the preceding string, except for the first two characters. Where the preceding string calls for a match to the prefix ******, this string redefines the inter-digit timeout. This value has been increased to eight (8) seconds. This timeout value persists until the first digit plus five other digits have been collected, at which time the timeout value is reduced to three (3) seconds. From that point onward, the VoIP Subsystem continues to collect digits until the user pauses three seconds, at which point the VoIP Subsystem sends the dialed string to the service provider.

#[1-9]e#r5xp3r*x is a string that is identical to the previous two, except for the first digit. This string supports cases where service providers use strings that start with # for various special features or control purposes. This string is forwarded to the service provider only if the mode is set to **Passthrough**.

1010Se#p2r*x is a string included to support cases where North-American style dial-around dialing is available. The **S** means that if someone dials 1010 as the first four digits of a dial string, this is the only string the VoIP Subsystem should match to from that point on. **e#** means that the user can indicate the completion of dialing at any time by entering **#. p2** means that after someone dials 1010, the timeout between subsequent digits is reduced to two (2) seconds. **r*x** means that the VoIP Subsystem will continue to collect dialed digits until there is a timeout.

0Se#r5xp2r*x is the second workhorse string of the default pattern matching string. International calls in almost every country, and domestic long distance calls in most countries outside North America, all match this pattern. Any number that starts with zero (**0**) matches this string. The user may dial # at any time to indicate the number dialed is complete. After the user dials the sixth digit, the inter-digit timeout is reduced to two seconds. After that point, the VoIP Subsystem continues to collect digits until the user pauses two seconds. Then the VoIP Subsystem sends the dialed string to the service provider.

[3469]11 means either **3** OR **4** OR **6** OR **9, followed by 11** (that is, **311** OR **411** OR **611** OR **911**).

North American Number Plan Area (NANPA) Dialing Examples

[^1]r6x

Recognize a seven (7) digit number, However, do not match to this string if beginning with a **1**(one)

This string will allow a user to dial **2XXXXXX - 9XXXXXX**. However, if the number entered begins with a **1** (one), do not match to this pattern.

1r3x[^1]r6x

Match a long distance number to this string, as in 1-<area code>-<7 digit dial>.

This string will allow a user to dial a phone number using a toll prefix of **1** (one). It also makes certain that the seven-digit local phone number under NANPA does not begin with a **1** (one).

Dial String Tips

1900r7x!

Disallow **1900XXXXXXX**

This tells the system to look at the first four digits of the entered number, and if they match **1900** to drop to a failed tone.

1900 numbers in the US are premium-rate numbers that may incur high per-minute charges.

976r4!

Disallow a **976XXXX** number

This tells the system to look at the first three digits of the entered number, and if they match **976** to drop to a failed tone.

. 976 numbers in the US are premium-rate local numbers that may incur high per-minute charges.

1800r7x

Recognize a **1800XXXXXXX** number

This tells the system to look at the first four digits of the entered number, and if they match **1800** to dial using **1800** plus the remaining seven digits.

<:>

If you want to set up a dial pattern that allows the user to easily select between two services, you can use the **<:>** symbol. By including **<[89]:>** in the dial pattern, you tell the system to replace an **8** or **9** with a null value, and continue pattern matching as necessary.

For example, **<[89]:>r7x:** as long as the first digit is an **8** or **9**, the system will accept an 8 or 9 followed by seven digits, remove the first digit (8 or 9), and dial out the remaining seven digits. You can specifiy an **8** as part of the pattern recognition string for one provider, and **9** as part of the pattern recognition for another provider. This will allow users to easily select among providers with similar numbers. Note that this doesn't work well if any numbers you want to reach start with **8** or **9**. In that case, you may want to consider prefixes that start with ***8, #8, *9** or **#9**.

Entering Easily-Confused Patterns

If you enter two different patterns which can easily be confused with each other, the system will choose the first pattern that is matched. For instance, if you have two patterns, one for eleven digits, and one for twelve, the system will not wait for the twelfth digit, because it will match to the eleven-digit pattern first. To prevent this, you should set up the dial pattern (matching similarly to the two examples above) using **0Se#e*p2r*x** or **1010Se#e*p2r*x.** These patterns will force the system to wait until after the user has entered as many digits as are necessary before it tries to connect to a provider.

Bridging From VoIP to PSTN

Parameter	Description	Default
Bridge from VoIP to PSTN	Enable or disable the bridge	Disabled
Auto-Answer VoIP Bridge Calls	Enable or disable auto-answer	Disabled
VoIP Bridge Accept Any Call	Enable or disable call acceptance	Disabled
VoIP Bridge Accept Anonymous Calls	Enable or disable anonymous call acceptance	Disabled
VoIP Bridge Single Stage Dialing Enable	Enable or disable single stage dialing	Disabled
Caller Password	Enable or disable caller password	Disabled
Password Dial String	Specifies the password dial string	
VoIP Bridge Accept Only These Numbers (01 to 10)	When any numbers are listed here, only calls from those numbers will be bridged.	
VoIP Bridge Billing Delay Duration (10 ms)	Specifies the duration of billing delay (0 to 65535 ms)	100 ms
VoIP Bridge Security Entry Duration (10 ms)	Specifies the duration for the security entry (0 to 65535 ms)	1000 ms

Bridging from PSTN to VoIP

Parameter	Description	Default
Bridge From PSTN to VoIP	Enable or disable the bridge	Disabled
Auto Answer PSTN (FXO) Calls	Enable or disable auto-answer PSTN calls	Disabled
FXO Port Accept Anonymous Calls	Enable or disable anonymous call acceptance on FXO port	Disabled
FXO Port Only Accept Calls with Caller ID	Enable or disable acceptance of caller ID calls only on FXO port	Disabled
FXO Port Accept Only These Numbers (01 to 10)	When any numbers are listed, only calls to those numbers will be accepted.	
Caller Password	Specifies requirement for caller password	Disabled
Password Dial String	Specifies required caller password string	

Miscellaneous TELCO Parameters

Parameter	Description	Default
Telco Port Display Caller ID	Enable or disable the caller ID display	Disabled
Telco Port Caller ID Sent After One Ring	Indicate to device whether Telco CID is sent before or after the first ring	Enabled
PSTN CID Wait Duration (10 ms)	Time after incoming call initiation (first ring or line reversal to continue looking for CID signal). (0 to 65535 ms)	500 ms
PSTN CID Clear Duration (10 ms)	Time after last ring to continue to display CID. (0 to 65535 ms)	1000 ms
Billing Delay Duration (10 ms)	Time after auto-answer to send Bong tone prompt in bridge mode. (0 to 65535 ms)	100 ms
PSTN Security Entry Duration (10 ms)	In bridge mode, time within which the user must enter security code, if enabled. (0 to 65535 ms)	1000 ms
If My Call Starts With These Digits	Requests the line to use when dialing numbers that begin with the specified digits.	
If I Normally Want Auto-Add Area Code Calls Routed	Enables or disables alternate auto-add routing of Telco line calls	Disabled
Route VoIP Calls Via My Telco Line If VoIP Service is Unavailable	Enables or disables alternative routing of VoIP calls.	Enabled

Emergency Services and eServices Events

The emergency services numbers follow the same rules as those defined for the pattern matching strings in Dialing Parameters on page 36.

The VoIP Subsystem allows flexible treatment of emergency numbers. They can be sent either via the Internet or over the PSTN. When you are connected to a SUBSCRIPTION server that supports the **Eservices** (Emergency Services) event, the server and VoIP Subsystem can coordinate with each other to make sure that the VoIP Subsystem will route emergency calls via the appropriate connection. Make sure to include all emergency numbers in both the default VoIP and PSTN parameters, if you want the VoIP Subsystem to make a flexible selection.

Parameter	Description	Default
Emergency Numbers Routed via VoIP	Specifies which emergency numbers to route over VoIP	100, 11x, 911, 999
Emergency Numbers Routed via the PSTN	Specifies which emergency numbers to route over PSTN	100, 11x, 911, 999
Default Emergency Numbers Routed via VoIP	Specifies which default emergency numbers to route over VoIP	100|11x|911|999
Default Emergency Numbers Routed via the PSTN	Specifies which default emergency numbers to route over PSTN	100|11x|911|999
Always Route Emergency Numbers via the PSTN	When enabled, this parameter configures the VoIP Subsystem to always send emergency numbers to the PSTN. If the PSTN line is unavailable, then emergency calls are routed via VoIP.	Disabled
Emergency Numbers via the PSTN Alt (Click Help)	When enabled, this parameter allows the VoIP Subsystem to determine which port to send emergency numbers to, based on negotiation over the event **Eservices** with the subscription server. If the subscription to the **Eservices** event fails, then emergency numbers are routed to the PSTN. If both **Always Route Emergency Numbers via PSTN** and **Emergency Numbers via the PSTN Alt** are both disabled, then emergency calls will be routed according to negotiation through the event **Eservices**. If the subscription fails, then emergency calls are preferentially routed via VoIP.	Enabled

Note: If neither the PSTN nor VoIP is available, users will hear no dial tone when they pick up the handset. In that case, they should understand that they cannot make an emergency call.

9
User Configuration

You can use the **VoIP > Advanced VoIP Settings > User Configuration** menu to configure the VoIP Subsystem's user-specific settings. The menu items include:

- Speed Dials
- Call Forwarding
- Ringing Based on Caller ID
- Do Not Disturb
- Incoming Call Blocking
- Call Waiting/Caller ID
- Timers

Speed Dials

The **Speed Dial List** can be modified by the telephone or via the web pages. Up to 28 numbers can be entered into the **Speed Dial List**. Each number can be up to 40 digits in length. Dialing a speed dial number is explained in Chapter 4 of the *Zoom ADSL X6v User Guide* on your X6v CD.

Parameter	Description	Default
***20 - *39**	Speed dial number corresponding to *20 to *39	
#0 - #7	Speed dial number corresponding to #0 to #7	

Call Forwarding

With **Call Forward** enabled, any call on this list will be forwarded to the number stored in the **Call Forward List** (1-12). Up to thirty 40-digit numbers can be entered.

Parameter	Description	Default
Call Forward Always	Enable or disable call forwarding in all cases	Disabled
Call Forward on Busy	Enable or disable call forwarding when line is busy	Disabled
Call Forward on No Answer	Enable or disable call forwarding when the call is not answered	Disabled

Parameter	Description	Default
Call Forward Priority	Enables or disables priority call forward	Disabled
Call Forward Always Number	Specifies the call forward destination	
Call Forward on Busy Number	Specifies the call forward destination when the line is busy	
Call Forward on No Ans Number	Specifies the call forward destination when the line is not answered	
Call Forward Priority Number	Specifies the priority call forward destination	
Priority Forward List – 1 to 30 phone numbers	Specifies the list of numbers	

Ringing Based on Caller ID

Parameter	Description	Default
Ringing Based on Caller ID	Enables or disables distinctive ring tones linked to caller IDs	Enabled
Distinctive Ring List – 1 to 30 phone numbers	Specifies the phone numbers associated with caller IDs	

Do Not Disturb

Parameter	Description	Default
Do Not Disturb Mode	Enables or disables the **Do Not Disturb Mode**, which blocks all non-priority calls. Priority calls are permitted if further enabled by the **Do Not Disturb Exceptions.** This value is reset on power up and restart.	Disabled
Do Not Disturb Exceptions	Enables or disables the ringing of calls on the **Disturb Exceptions List**. All other callers will be blocked.	Disabled
Do Not Disturb Exceptions List – 1 to 30 phone numbers	Specifies the list of numbers	

Incoming Call Blocking

Parameter	Description	Default
Block Anonymous Incoming Calls	Enables or disables the blocking of calls that do not give caller ID information	Disabled
Block Listed Incoming Calls	Enables or disables the blocking of incoming calls from specific numbers in the **Blocked Call List**	Disabled
Blocked Call List – 01 to 30 numbers	Specifies the list of incoming numbers	

Parameter	Description	Default
Block Listed Outgoing Calls	Enables or disables the blocking of outgoing calls from specific numbers in the **Blocked Call List**	Disabled
Blocked Call List – 01 to 30 numbers	Specifies the list of outgoing numbers	

Call Waiting/Caller ID

Availability of these features depends on whether they are supported by your VoIP service provider.

Parameter	Description	Default
Call Waiting	Enables or disables call waiting for all calls. When the line is in use and a call is received, a call waiting tone is played. Pressing the flash or the hook button on the phone momentarily switches between the two calls. While there are calls on both lines, additional incoming calls receive busy signals.	Enabled
Inbound Caller ID	Enables or disables caller ID for inbound calls	Enabled
Outbound Caller ID	Enables or disables caller ID for outbound calls	Enabled
Call Waiting Caller ID	Enables or disables caller ID during call waiting	Enabled

Timers

Parameter	Description	Default
Brief pause (10 ms)	Sets the amount of time after picking up the receiver before dial tone is generated. (The range is 0 to 65535 in units of 10 ms)	50 (that is, 500 ms)
Initial Dial (10 ms)	Specifies amount of time allowed for the user to dial a digit after picking up the telephone receiver. (The range is 0 to 65535 in units of 10 ms)	1500 (15 s)
Warm Line (10 ms)	Specifies the amount of time from when the receiver is picked up to the first dialed digit before Warm Line dialing occurs. (The range is 0 to 65535 in units of 10 ms)	400 (4 s)

Parameter	Description	Default
Interdigit (10 ms)	Specifies the amount of time the VoIP Subsystem waits after the dial string has matched a dial pattern. After this amount of time, the VoIP Subsystem will go ahead and dial that number. (The range is 0 to 65535 in units of 10 ms)	500 (5 s)
Dialing (10 ms)	Specifies the amount of time between digits before a timeout occurs. This may be overridden by the ‘p’ parameter in a Dial String. (The range is 0 to 65535 in units of 10 ms)	1000 (10 s)
Hangup Disconnect (10 ms)	Specifies the amount of time to wait (after the disconnect command) before transitioning to the standby state. (The range is 0 to 65535 in units of 10 ms)	85 (850 ms)
Hangup Silence (10 ms)	Used if **Hangup Disconnect** is not enabled; that is, does not have a value. (The range is 0 to 65535 in units of 10 ms)	1000 (10 s)
No Answer (s)	Relative to call forwarding -- time after which a call-waiting call is considered to be a No Answer call. After this time the call will be forwarded if Forward on No Answer is enabled. (The range is 0 to 65535 s)	20 s
Pause Wait (10 ms)	Time that device will pause when a pause symbol is entered in a string that will be dialed onto the PSTN via the FXO port. (The range is 0 to 65535 in units of 10 ms)	300 (3 s)
Timeout Tone (10 ms)	If a timeout occurs during dialing or answering, a busy signal is sent to the telephone. The dialing duration specifies the amount of time to send the busy signal. (The range is 0 to 65535 in units of 10 ms)	1000 (10 s)
Timeout Pause (10 ms)	Specifies the amount of time between the busy and alert tones. (The range is 0 to 65535 in units of 10 ms)	100 (1 s)

Parameter	Description	Default
Timeout Disconnect (10 ms)	The range is 0 to 65535 in units of 10 ms	85 (850 ms)
Timeout Warning (10 s)	When the telephone is off hook for too long, the alert tone is sent to the phone. The amount of time for the alert tone is specified by the alert duration. (The range is 0 to 65535 s)	1 (10 s)
Timeout Hold (10 ms)	When a call is placed on hold, this parameter specifies the amount of time to wait before the call holding tone is played. (The range is 0 to 65535 in units of 10 ms)	1000 (10 s)
Timeout Hold Drop (10 ms)	Drop a call on hold after this time. (The range is 0 to 65535 in units of 10 ms)	6000 (60 s)
Timeout No Answer Drop (s)	If forwarding is not enabled, an incoming call-waiting call is dropped after the specified amount of time. (The range is 0 to 65535 ms)	120 s
Call Back (s)	Not implemented.	
Call Back Retry (s)	Not implemented.	
Call Back Ring Wait (s)	Not implemented.	
Message Waiting Refresh (s)	Request updates to voice message status at this interval.	1800 (30 min)
Hookflash Maximum (ms)	Sets the maximum amount of time for the telephone receiver to stay on-hook before it is regarded as simply on-hook. If the receiver is on-hook for more than the minimum hook-flash time and less than the maximum hook-flash time, the system recognizes hook-flash. (The range is 0 to 1600 ms.)	900 ms
Hookflash Minimum (ms)	Sets the minimum amount of time for the telephone receiver to stay on-hook in order to be regarded as hook-flash. If the receiver does not stay on-hook for the hookflash minimum time, the VoIP Subsystem does not recognize hookflash as having occurred. (The range is 0 to 4150 ms.)	300 ms
Hookflash Delay (ms)	The range is 0 to 1000 ms	200 ms
Answer Hangup Delay (ms)	Sets the minimum amount of time for the telephone receiver to stay on-hook before the VoIP Subsystem ends the current call. This applies only to incoming calls. (The range is 0 to 60,000 ms)	0 ms

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Feature Codes

Feature codes are used to access advanced Class 5 telephony features. You can use the **VoIP -> Advanced VoIP Settings -> Feature Codes** menu to configure the parameters. The menu includes:

- Feature Code Assignments (*55 – *99)

Feature Code Assignments (*55 – *99)

The IPBX calling features are assigned the ranges *55 to *89 and *92 to *99. The codes can be re-assigned to better match common local conventions, but they must be given codes within the assigned ranges. The default values represent the commonly used assignments.

Parameter	Description	Default
Call Waiting Enable	Enable call waiting on all calls	*55
Call Waiting Disable	Disable call waiting on all calls	*56
Call Trace	Call trace (reserved)	*57
Call Waiting Caller ID Enable	Enable call waiting caller ID generation	*58
Call Waiting Caller ID Disable	Disable call waiting caller ID generation	*59
Blocked Number Enable	Enable call blocking feature	*60
Distinctive Ring Enable	Enable distinctive ringing feature	*61
Caller ID Outbound Disable	Block caller ID on all outbound calls	*62
Priority Forward Enable	Enable priority call forwarding feature	*63
Disturb Accept Enable	Enable do not disturb accept call feature	*64
Caller ID Inbound Enable	Enable caller ID generation	*65
Busy Number Redial	Busy number redial	*66
Caller ID Outbound One-time Enable	Unblock caller ID for one call	*67
Caller ID Outbound One-time Disable	Block caller ID for one call	*68
Caller Redial	Call the last caller	*69
Call Waiting One-time Disable	Deactivate call waiting for current call	*70
Call Waiting One-time Enable	Enable call waiting for current call	*71
Call Forward Enable	Enable call forwarding to number that follows	*72
Call Forward Disable	Cancel call forwarding of non-priority calls	*73
One Digit Speed Dial Program	Program speed dials 0 - 7	*74
Two Digit Speed Dial Program	Program speed dials 20 - 39	*75
Block Anonymous Enable	Block all anonymous calls	*77

Parameter	Description	Default
Do Not Disturb Enable	Enter do not disturb state	*78
Do Not Disturb Disable	Exit do no disturb state	*79
Blocked Number Disable	Cancel call lock - remove optional number from blocked call list, or disable call blocking	*80
Distinctive Ring Disable	Disable distinctive ringing	*81
Caller ID Outbound Enable	Unblock caller ID on all outbound calls	*82
Priority Forward Disable	Cancel priority call forward	*83
Disturb Accept Disable	Disable do not disturb accept call feature	*84
Caller ID Inbound Disable	Disable caller ID generation	*85
Busy Number Redial Cancel	Cancel busy redial	*86
Block Anonymous Disable	Unblock anonymous calls	*87
Caller Redial Cancel	Cancel calling last caller	*89
Forward No Answer Enable	Call forward when no answer - number follows	*92
Forward No Answer Disable	Cancel call forward when no answer	*93
Forward Busy Enable	Call forward when busy - number follows	*94
Forward Busy Disable	Cancel call forward when busy	*95
Outgoing Block Enable	Enable Block Outgoing VoIP calls feature	*96
Outgoing Block Disable	Disable Block Outgoing VoIP calls feature	*97
Unattended Transfer	Execute Hook Flash followed by *98 to initiate unattended transfer	*98

NOTICE